AMENDMENTS TO THE CLAIMS:

This listing of claims will replace all prior versions and listings of claims in the application:

LISTING OF CLAIMS:

1. (Original) An apparatus for enhancing speech quality in digital communications, comprising:

an input buffer for storing a sum signal of a first input signal to be transmitted and an echo signal generated from a received second input signal at a predetermined time interval;

an echo canceller for receiving the sum signal based on a unit of a buffer from the input buffer, canceling the echo signal from the sum signal, and outputting the first input signal;

a noise canceller for receiving the first input signal based on the buffer unit from the echo canceller, and canceling a noise from the first input signal;

a level controller for receiving the first input signal based on the buffer unit from the noise canceller, and adjusting a level of the first input signal; and

a speech compression module for receiving the first input signal based on the buffer unit from the level controller, converting the first input signal into a digital signal, and compressing the digital signal.

2. (Original) The apparatus as set forth in claim 1, wherein the speech compression module converts the first input signal into the digital signal, compresses the digital signal at a variable rate, determines whether a speech signal exists within the first input signal in a compression operation, generates speech-signal determination information, and outputs the speech-signal determination information to the echo canceller.

3. (Original) The apparatus as set forth in claim 1, wherein the speech compression module generates characteristic information associated with the first input signal in an operation of converting the first input signal into the digital signal, and outputs the characteristic information to the echo canceller.

4. (Original) The apparatus as set forth in claim 3,

wherein the speech compression module comprises an LPC (Linear Prediction Coding) analyzer, pitch analyzer and codebook analyzer associated with the first input signal, and operates on the basis of CELP (Code Excited Linear Prediction), and

wherein the characteristic information is configured by at least one of LPC information, pitch information and codebook information associated with the first input signal.

5. (Original) The apparatus as set forth in claim 4, wherein the speech compression module calculates a parameter quantization error associated with the characteristic information, calculate an error between the first input signal and a recovered first input signal to generate speech compression performance information, and outputs the speech compression performance information to at least one of the noise canceller and level controller.

6. (Original) The apparatus as set forth in claim 5, wherein the noise canceller comprises:

a band-by-band noise estimator for estimating a frequency band-by-band noise of the first input signal; and

a noise component subtracter for subtracting a noise component estimated by the band-by-band noise estimator from a frequency band-by-band signal of the first input signal,

wherein at least one of the noise component subtracter and band-by-band noise estimator receives the speech compression performance information and performs an operation of allowing the speech compression module to increase a compression performance.

7. (Original) The apparatus as set forth in claim 5, wherein the level controller comprises:

a level estimator for receiving the first input signal and

estimating a speech signal level;

a level conversion decider for deciding a conversion level with the level estimated by the level estimator; and

a level converter for converting the speech signal level into the level decided by the level conversion decider,

wherein the level estimator, level conversion decider and level converter receive the speech compression performance information and determine level adjustment performance associated with the speech signal.

8. (Original) The apparatus as set forth in claim 1, further comprising:

a speech decompression module for receiving the digital signal and decompressing the digital signal into the second input signal; and

an output buffer for storing the second input signal at a predetermined time interval, the stored second input signal based on the buffer unit being outputted to the echo canceller.

9. (Original) The apparatus as set forth in claim 8, wherein the speech decompression module decompresses the digital signal into the second input signal at a variable rate, determines whether or not a speech signal exists within the second input signal in an operation of decompressing the digital signal into

the second input signal, generates speech-signal determination information, and outputs the speech-signal determination information to the echo canceller.

10. (Currently Amended) The apparatus as set forth in claim 2 ~~or 9~~, wherein the echo canceller comprises:

a DT (Double-Talk) detector for determining whether a speech signal exists within the first and second input signals;

an adaptive filter for predicting the echo signal according to a result of the determination from the DT detector;

an operator for producing a difference signal between the sum signal and the echo signal predicted from the adaptive filter; and

a non-linear processor for finally canceling a remaining echo signal from the difference signal,

wherein at least one of the DT detector and the adaptive filter receives the speech-signal determination information and uses the received speech-signal determination information as the result of the determination by the DT detector.

11. (Original) The apparatus as set forth in claim 8, wherein the speech decompression module outputs, to the echo canceller, characteristic information associated with the second input signal generated in an operation of decompressing the digital signal into

the second input signal.

12. (Original) The apparatus as set forth in claim 11, wherein the speech decompression module comprises an LPC (Linear Prediction Coding) synthesizer, pitch synthesizer and codebook synthesizer associated with the second input signal, and operates on the basis of CELP (Code Excited Linear Prediction), and

wherein the characteristic information is configured by at least one of LPC information, pitch information and codebook information associated with the second input signal.

13. (Currently Amended) The apparatus as set forth in claim 4 ~~or 12~~, wherein the echo canceller comprises:

a DT (Double-Talk) detector for determining whether a speech signal exists within the first and second input signals;

an adaptive filter for predicting the echo signal according to a result of the determination from the DT detector;

an operator for producing a difference signal between the sum signal and the echo signal predicted from the adaptive filter; and

a non-linear processor for finally canceling a remaining echo signal from the difference signal,

wherein at least one of the DT detector and the adaptive filter receives the speech-signal determination information and

uses the received speech-signal determination information as additional information in the determination by the DT detector.

14. (Original) The apparatus as set forth in claim 8, wherein the echo canceller comprises:

a DT (Double-Talk) detector for determining whether a speech signal exists within the first and second input signals;

an adaptive filter for predicting the echo signal according to a result of the determination from the DT detector;

an operator for producing a difference signal between the sum signal and the predicted echo signal from the adaptive filter; and

a non-linear processor for finally canceling a remaining echo signal from the difference signal,

wherein the result of the determination from the DT detector is outputted into at least one of the noise canceller and level controller.

15. (Currently Amended) The apparatus as set forth in claim 1 ~~or 8~~, wherein the predetermined time interval is within a range of 10 msec to 30 msec.

16. (Original) A method for enhancing speech quality in digital communications, comprising the steps of:

(a) storing a sum signal of a first input signal to be

remotely transmitted and an echo signal generated from a remotely received second input signal at a predetermined time interval;

(b) receiving the sum signal based on a unit of a buffer, canceling the echo signal from the sum signal, and extracting the first input signal;

(c) receiving the first input signal based on the buffer unit, and canceling a noise from the first input signal;

(d) receiving the first input signal based on the buffer unit in which the noise is cancelled, and adjusting a level of the first input signal; and

(e) receiving the first input signal based on the buffer unit in which the level of the first input signal is adjusted, converting the first input signal into a digital signal, and compressing the digital signal.

17. (Original) The method as set forth in claim 16, wherein the step (a) comprises the step of receiving the second input signal from the digital signal, recovering a speech signal from the second input signal, and buffering and outputting the speech signal at the predetermined time interval.

18. (Original) The method as set forth in claim 17, wherein the step (a) comprises the step of generating characteristic information associated with the second input signal in an

operation of recovering the speech signal from the second input signal, and

the step (e) comprises the step of generating characteristic information associated with the first input signal in an operation of converting the first input signal into the digital signal and compressing the digital signal.

19. (Original) The method as set forth in claim 18,

wherein at the step (a) the characteristic information is configured by at least one of LPC (Linear Prediction Coding) information, pitch information and codebook information associated with the second input signal, and

wherein at the step (e) the characteristic information is configured by at least one of LPC information, pitch information and codebook information associated with the first input signal.

20. (Original) The method as set forth in claim 18, wherein the step (b) comprises the step of determining whether a speech signal exists within the first and second input signals, predicting the echo signal according to a result of the determination, producing a difference signal by subtracting the predicted echo signal from the sum signal, and finally canceling a remaining echo signal from the difference signal,

wherein the characteristic information is received in an

operation of determining the existence of the speech signal, and is used as additional information.

21. (Original) The method as set forth in claim 16,

wherein the step (a) comprises the step of determining whether a speech signal exists within the second input signal in an operation of recovering the speech signal from the second input signal at a variable rate, and generating speech-signal determination information, and

wherein the step (e) comprises the step of determining whether a speech signal exists within the first input signal in an operation of converting the first input signal into the digital signal and compressing the digital signal at a variable rate, and generating speech-signal determination information.

22. (Original) The method as set forth in claim 19, wherein the step (e) comprises the step of calculating a parameter quantization error associated with the characteristic information, and calculating an error between the first input signal and a recovered first input signal to generate speech compression performance information.

23. (Original) The method as set forth in claim 22, wherein the step (c) comprises the step of receiving the speech

compression performance information to be used as additional information for estimating a frequency band-by-band noise of the first input signal, and canceling the noise from the first input signal so that speech compression performance can be enhanced.

24. (Original) The method as set forth in claim 22, wherein the step (d) comprises the step of receiving the speech compression performance information to be used as additional information for determining performance of adjusting a level of a speech signal contained in the first input signal.

25. (New) The apparatus as set forth in claim 9, wherein the echo canceller comprises:

a DT (Double-Talk) detector for determining whether a speech signal exists within the first and second input signals;

an adaptive filter for predicting the echo signal according to a result of the determination from the DT detector;

an operator for producing a difference signal between the sum signal and the echo signal predicted from the adaptive filter; and

a non-linear processor for finally canceling a remaining echo signal from the difference signal,

wherein at least one of the DT detector and the adaptive filter receives the speech-signal determination information and

uses the received speech-signal determination information as the result of the determination by the DT detector.

26. (New) The apparatus as set forth in claim 12, wherein the echo canceller comprises:

a DT (Double-Talk) detector for determining whether a speech signal exists within the first and second input signals;

an adaptive filter for predicting the echo signal according to a result of the determination from the DT detector;

an operator for producing a difference signal between the sum signal and the echo signal predicted from the adaptive filter; and

a non-linear processor for finally canceling a remaining echo signal from the difference signal,

wherein at least one of the DT detector and the adaptive filter receives the speech-signal determination information and uses the received speech-signal determination information as additional information in the determination by the DT detector.

27. (New) The apparatus as set forth in claim 8, wherein the predetermined time interval is within a range of 10 msec to 30 msec.